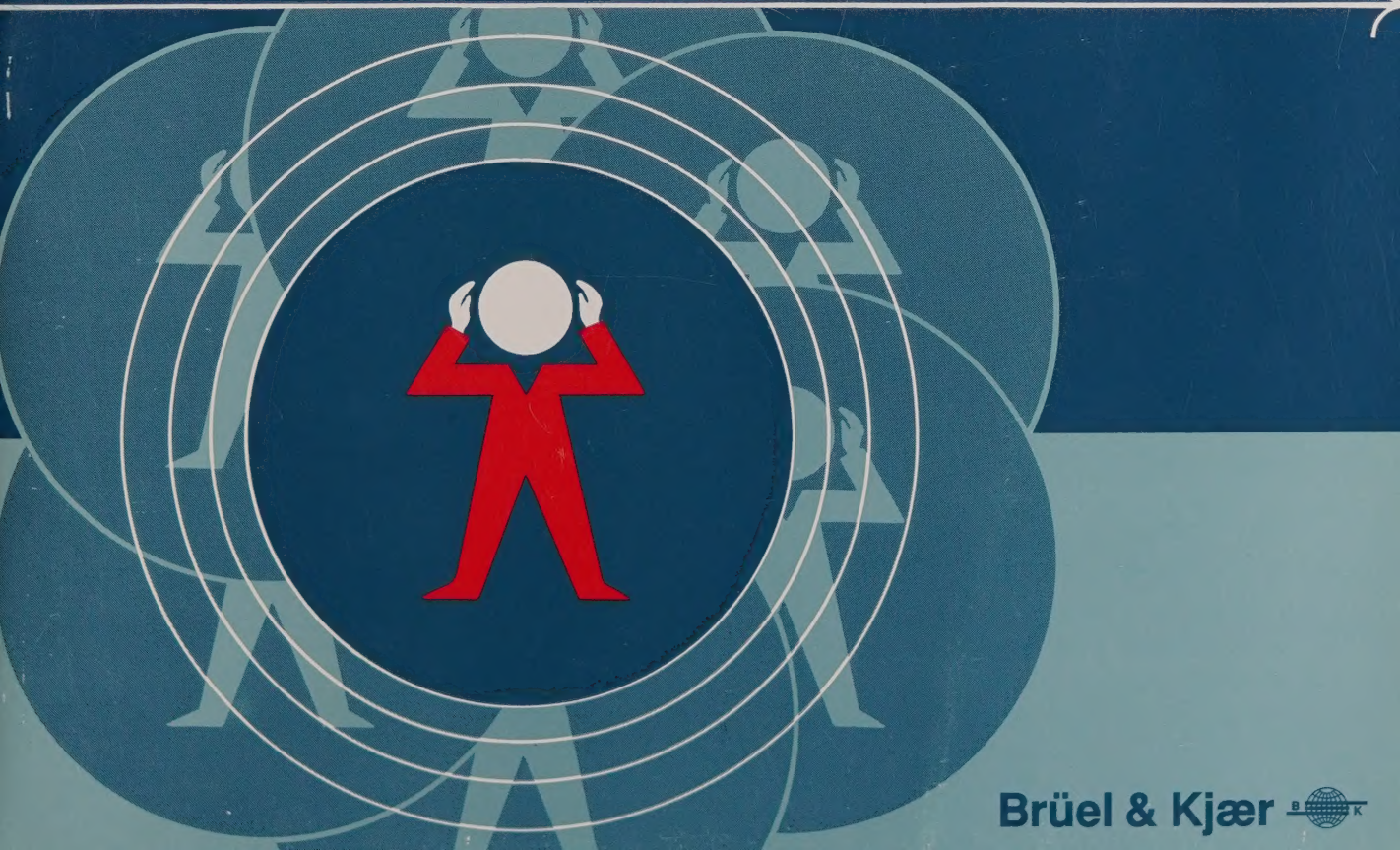
Measuring Sound



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Introduction

This booklet gives answers to some of the basic questions asked by the newcomer to a noise measuring programme. It gives a brief explanation to questions like:

What is sound ?

Why do we measure sound ?

What units do we use ?

How do we hear ?

What instruments do we use for measurement ?

What is a weighting network ?

What is frequency analysis ?

What is noise dose ?

How does sound propagate ?

Where should we make our measurements ?

How does the environment influence measurements ?

How should the microphone be positioned in the sound field ?

How do we make a measurement report ?

What do we do when levels are too high ?

Sound and the Human Being

Sound is such a common part of everyday life that we rarely appreciate all of its functions. It provides enjoyable experiences such as listening to music or to the singing of birds. It enables spoken communication with family and friends. It can alert or warn us — for example with the ringing of a telephone, or a wailing siren. Sound also permits us to make quality evaluations and diagnoses — the chattering valves of a car, a squeaking wheel, or a heart murmur.

Yet, too often in our modern society, sound annoys us. Many sounds are unpleasant or unwanted — these are called **noise.** However, the level of annoyance depends not only on the quality of the sound, but also our attitude towards it. The sound of his new jet aircraft taking off may be music to the ears of the design engineer, but will be ear-splitting agony for the people living near the end of the runway. But sound doesn't need to be loud to annoy. A creaking floor, a scratch on a record, or a dripping tap can be just as annoying as loud thunder.

Worst of all, sound can damage and destroy. A sonic boom can shatter windows and shake plaster off walls. But the most unfortunate case is when sound damages the delicate mechanism designed to receive it — the human ear.

Why Measure Sound ?

Measurements provide definite quantities which describe and rate sounds. These measurements can provide benefits such as improved building acoustics and loudspeakers, thus increasing our enjoyment of music, both in the concert hall and at home.

Sound measurements also permit precise, scientific analysis of annoying sounds. However, we must remember that due to the physiological and psychological differences between individuals, the degree of annoyance cannot be scientifically measured for a given person. But the measurements do give us an objective means of comparing annoying sounds under different conditions.

Sound measurements also give a clear indication of when a sound may cause damage to hearing and permit corrective measures to be taken. The degree of hearing damage can be determined by **audiometry** which measures a person's hearing sensitivity. Thus, sound measurements are a vital part of hearing conservation programmes.

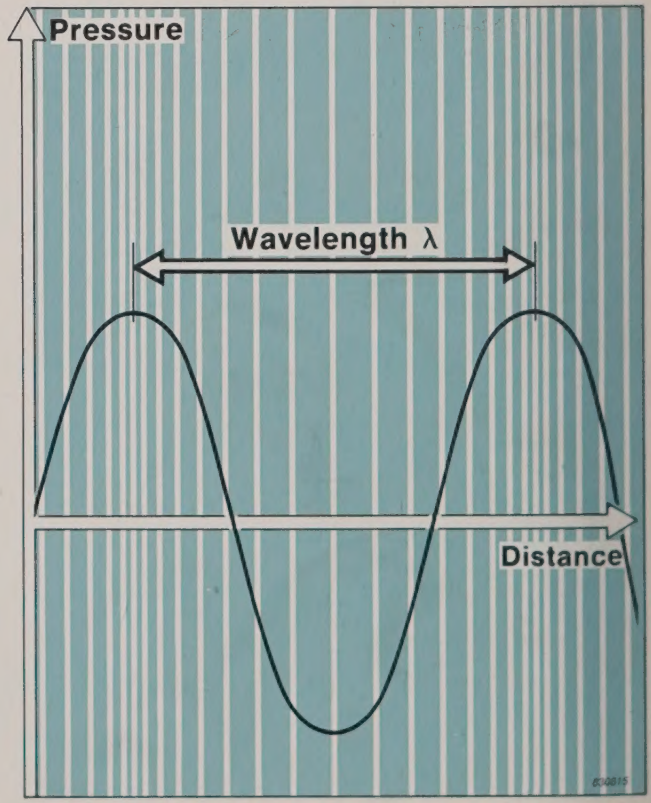
Finally, measurement and analysis of sound is a powerful diagnostic tool in noise reduction programmes — from airports, to factories, highways, homes and recording studios. It is a tool which can help to improve the quality of our lives.

Just What is Sound ?

Sound may be defined as **any pressure variation** (in air, water or other medium) **that the human ear can detect**. The most familiar instrument for measuring pressure variations in air is the barometer. However, the pressure variations which occur with changing weather conditions are much too slow for the human ear to detect — and hence do not meet our definition of sound. But, if variations in atmospheric pressure occur more rapidly — at least 20 times a second — they can be heard and hence are called sound. (A barometer cannot respond quickly enough and therefore cannot be used to measure sound).

The number of pressure variations per second is called the **frequency** of the sound, and is measured in **Hertz (Hz)**. The frequency of a sound produces it's distinctive **tone**. Thus, the rumble of distant thunder has a low frequency, while a whistle has a high frequency. The normal range of hearing for a healthy young person extends from approximately 20 Hz up to 20 000 Hz (or 20 kHz) while the range from the lowest to highest note of a piano is 27,5 Hz to 4 186 Hz.

These pressure variations travel through any elastic medium (such as air) from the source of the sound to the listener's ears. You probably already have some idea of the speed of sound from the familiar rule for determining how far away a thunder storm is: count 3 seconds per kilometer or 5 seconds per mile from the time you see the lightning until you hear the thunder. This time interval corresponds to a speed of sound in air of 1238 km/hour or 770 miles per hour. For acoustic and sound measurement purposes, this speed is expressed as 344 meters per second at room temperature.

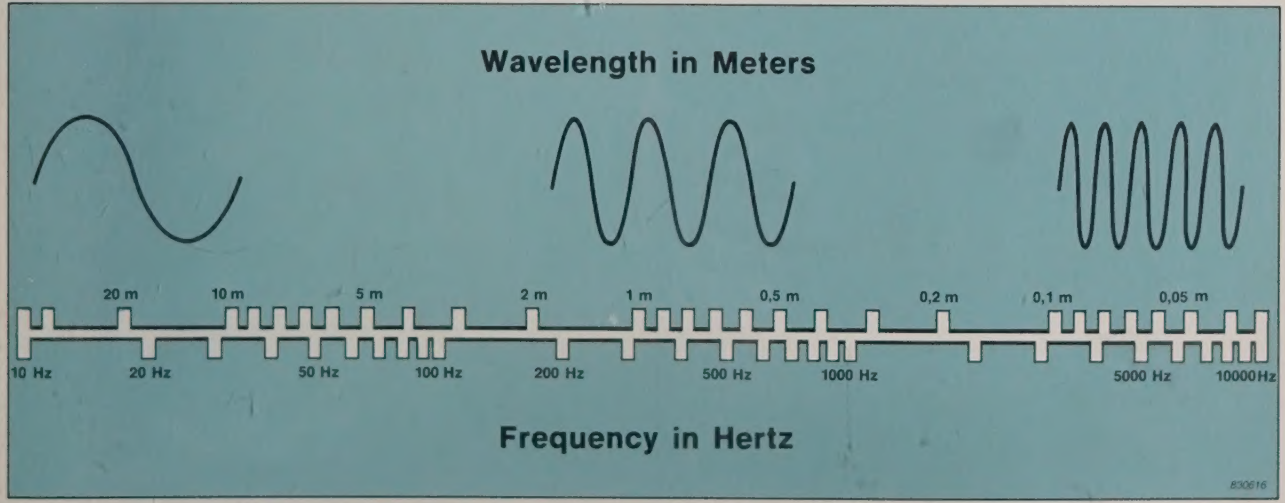


Knowing the speed and frequency of a sound, we can calculate the **wavelength** — that is, the distance from one wave top or pressure peak to the next.

$$\text{Wavelength } (\lambda) = \frac{\text{Speed of sound}}{\text{Frequency}}$$

From this equation we can work out the wavelength at different frequencies. For example at 20 Hz one wavelength is just over 17 meters, while at 20 kHz, it is only 1,7 cm. Thus, we see high frequency sounds have short wavelengths and low frequency sounds have long wavelengths.

A sound which has only one frequency is known as a **pure tone**. In practice pure tones are seldom encountered and most sounds are made up of different frequencies. Even a single note on a piano has a complex waveform. Most industrial noise consists of a wide mixture of frequencies known as **broad band noise.** If the noise has frequencies evenly distributed throughout the audible range it is known as **white noise** and it sounds rather like rushing water.

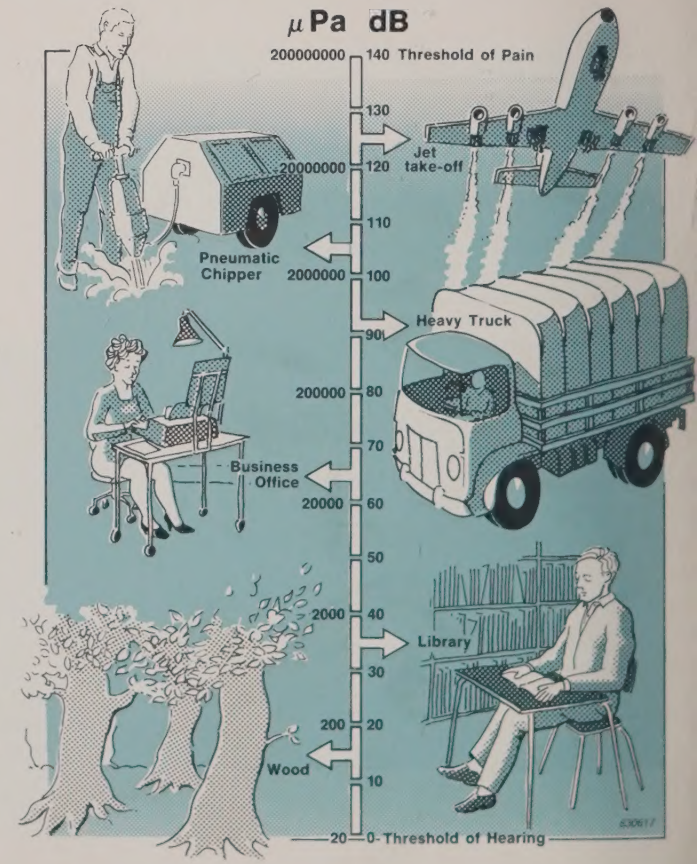


The dB

The second main quantity used to describe a sound is the size or **amplitude** of the pressure fluctuations. The weakest sound a healthy human ear can detect has an amplitude of 20 millionths of a Pascal (20 μPa) — some 5 000 000 000 times less than normal atmospheric pressure. A pressure change of 20 μPa is so small that it causes the eardrum to deflect a distance less than the diameter of a single hydrogen molecule. Amazingly, the ear can tolerate sound pressures more than a million times higher. Thus, if we measured sound in Pa, we would end up with some quite large, unmanageable numbers. To avoid this, another scale is used — the **decibel** or **dB scale**.

The decibel is not an absolute unit of measurement. It is a ratio between a measured quantity and an agreed reference level. The dB scale is **logarithmic** and uses the hearing threshold of 20 μPa as the reference level. This is defined as **0 dB**. When we **multiply** the sound pressure in Pa by 10, we **add** 20 dB to the dB level. So 200 μPa corresponds to 20 dB (re 20 μPa), 2 000 μPa to 40 dB and so on. Thus, the dB scale compresses a range of a million into a range of only 120 dB. The **sound pressure levels (SPL)** in dB and Pa of various familiar sounds are shown in the figure.

One useful aspect of the decibel scale is that it gives a much better approximation to the human perception of relative loudness than the Pascal scale. This is because the ear reacts to a logarithmic change in level, which corresponds to the decibel scale where 1 dB is the same relative change everywhere on the scale.

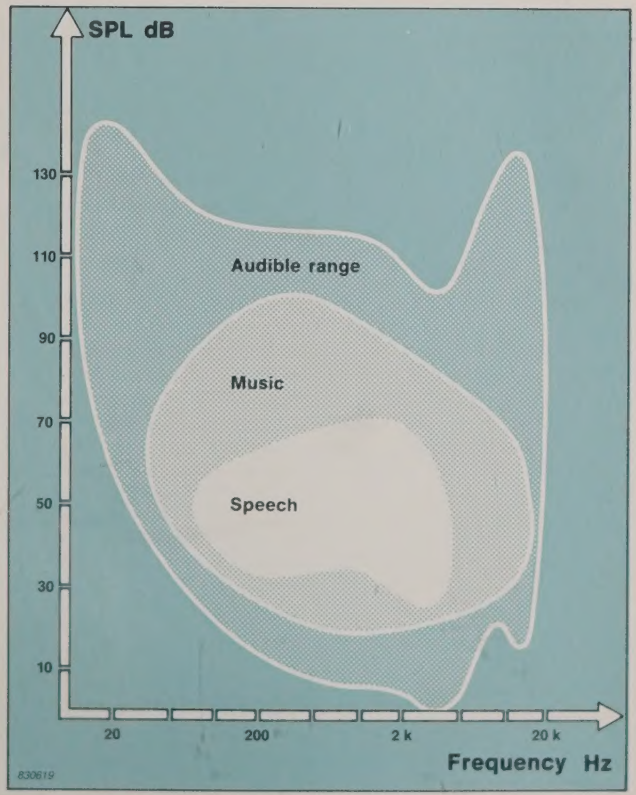


What do we Hear ?

We have already defined sound as any pressure variation which can be heard by a human ear. This means a range of frequencies from 20 Hz to 20 kHz for a young, healthy human ear. In terms of sound pressure level, audible sounds range from the **threshold of hearing** at 0 dB to the **threshold of pain** which can be over 130 dB.

Although an increase of 6 dB represents a doubling of the sound pressure, an increase of about 10 dB is required before the sound subjectively appears to be twice as loud. (The smallest change we can hear is about 3 dB).

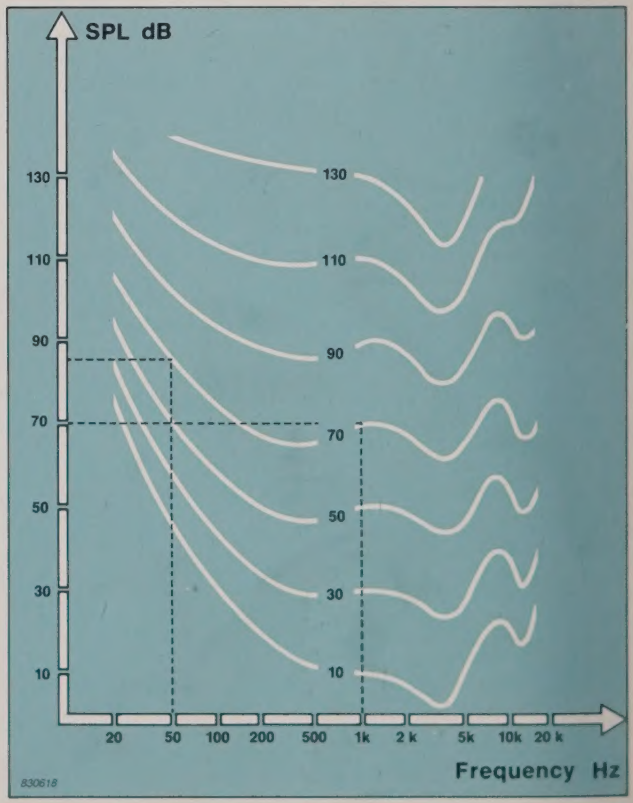
The subjective or **perceived** loudness of a sound is determined by several complex factors. One such factor is that the human ear is not equally sensitive at all frequencies. It is most sensitive to sounds between 2 kHz and 5 kHz, and less sensitive at higher and lower frequencies.



To complicate things even more, this difference in sensitivity to different frequencies is more pronounced at low SPLs than at high SPLs. This can be seen in the figure which shows a family of **equal loudness contours**. These indicate the sound pressure level required at any frequency in order to give the same apparent loudness as a 1 kHz tone.

For example, a 50 Hz tone must be 15 dB higher than a 1 kHz tone at a level of 70 dB in order to give the same subjective loudness.

Impulse sounds present another problem in loudness evaluation. If a sound is of short duration, that is, less than one second, it is termed an **impulsive** or **impulse** sound. Practical examples of impulse sounds are typewriter and hammering noises. Because of the short duration of such sounds, the ear is less sensitive perceiving it's loudness. Researchers generally agree that the perceived loudness of sounds shorter than 70 milliseconds (70 thousandths of one second) is less than that of sounds of longer durations having the same level.

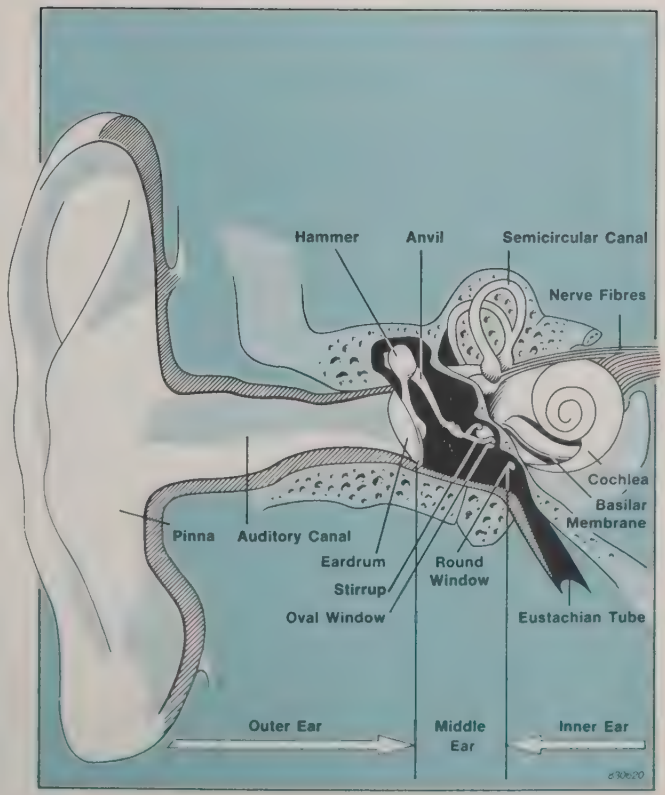


How do we Hear ?

The human ear consists of three main parts; the **outer ear, middle ear** and **inner ear.** The outer ear, consisting of the **pinna** and **auditory canal**, collects the airborne sound waves which then vibrate the **eardrum**, which is the interface with the middle ear. The middle ear acts as an impedance matching device and has three small bones operating as a set of levers. These bones transfer the vibration to the inner ear which consists of two separate systems, the **semi-circular canals** for controlling balance and the **cochlea.** The cochlea is a fluid-filled, snail-shaped tube which is divided longitudinally into two parts by the **basilar membrane**.

In response to an acoustic stimulus the fluid in the cochlea is disturbed and this distorts the basilar membrane on whose upper surface are thousands of very sensitive hair cells. The hair cells register this distortion and transform it into nerve impulses which are then transmitted to the brain.

Prolonged exposure to loud sounds causes damage to the hair cells with the result that hearing ability becomes progressively impaired. At first, damage to a few hair cells is not noticeable, but as more of the hair cells become damaged, the brain can no longer compensate for the loss of information. Words run into each other, speech and background noise cannot be distinguished and music becomes muffled. Considerable and irreparable damage will have occurred by the time the listener becomes aware of the loss. Loss of hearing caused by noise exposure is normally greatest at those frequencies (around 4 kHz) where the ear is most sensitive.

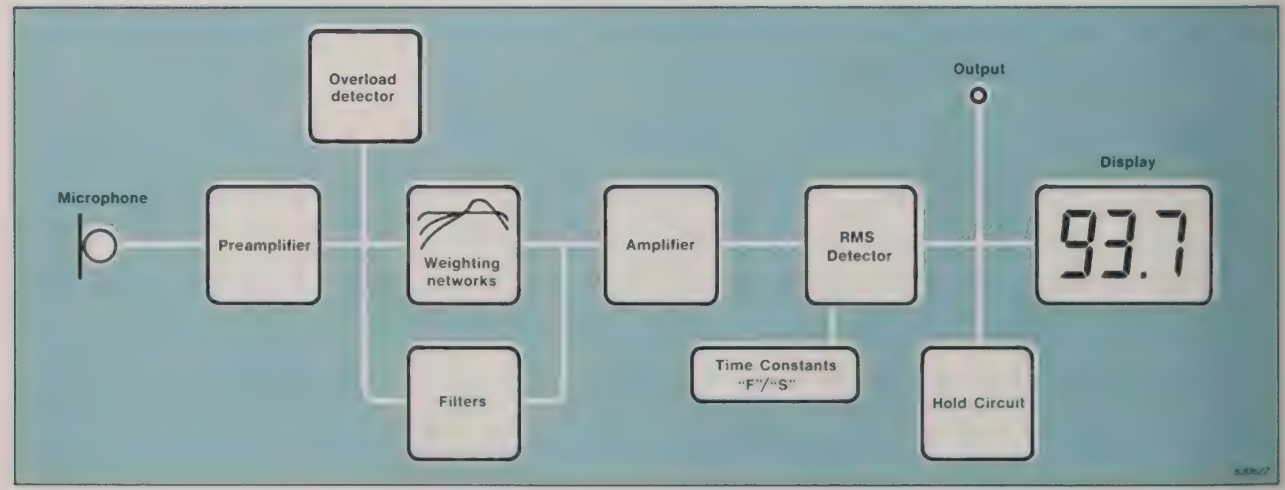


The Basic Sound Level Meter

A sound level meter is an instrument designed to respond to sound in approximately the same way as the human ear and to give objective, reproducible measurements of sound pressure level. There are many different sound measuring systems available. Although different in detail, each system consists of a microphone, an processing section and a read-out unit.

The microphone converts the sound signal to an equivalent electrical signal. The most suitable type of microphone for sound level meters is the **condenser microphone**, which combines precision with stability and reliability. The electrical signal produced by the microphone is quite small and so it is amplified by a preamplifier before being processed.

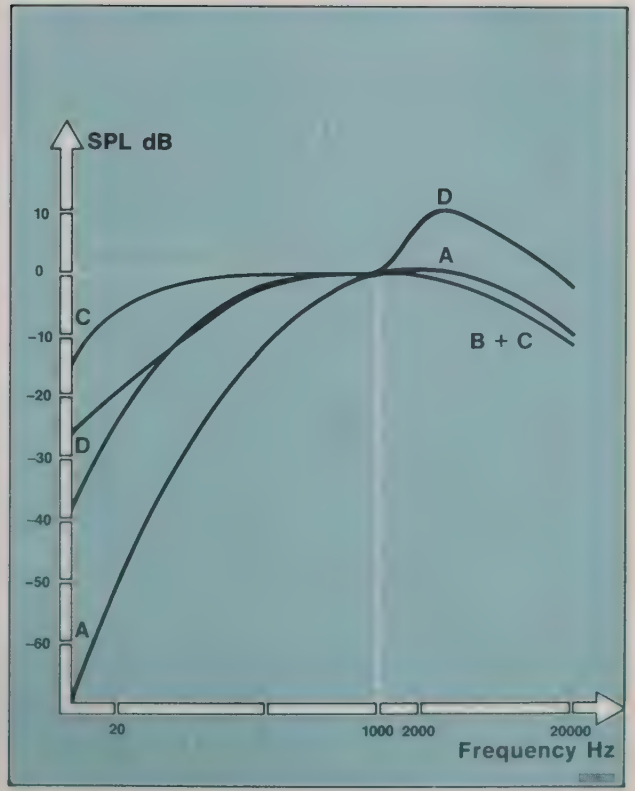
Several different types of processing may be performed on the signal. The signal may pass through a **weighting network**. It is relatively simple to build an electronic circuit whose sensitivity varies with frequency in the same way as the human ear, thus simulating the equal loudness contours. This has resulted in three different internationally standardized characteristics termed the **"A"**, **"B"** and **"C" weightings**.



The "A" weighting network weights a signal in a manner which approximates to an inverted equal loudness contour at low SPLs, the "B" network corresponds to a contour at medium SPLs and the "C" network to an equal loudness contour at high SPLs. A specialized characteristic, the **"D"** weighting, has also been standardized for aircraft noise measurements.

In addition to one or more of these weighting networks, sound level meters usually also have a **Linear** or **"Lin."** network. This does not weight the signal but enables the signal to pass through unmodified.

Nowdays the "A" weighting network is the most widely used since the "B" and "C" weightings do not correlate well with subjective tests. One reason for this lack of correlation between subjective tests and "B" and "C" weighted measurements is because the equal loudness contours were based on experiments which used **pure tones** — and most common sounds are not pure tones, but very complex signals made up of many different tones.

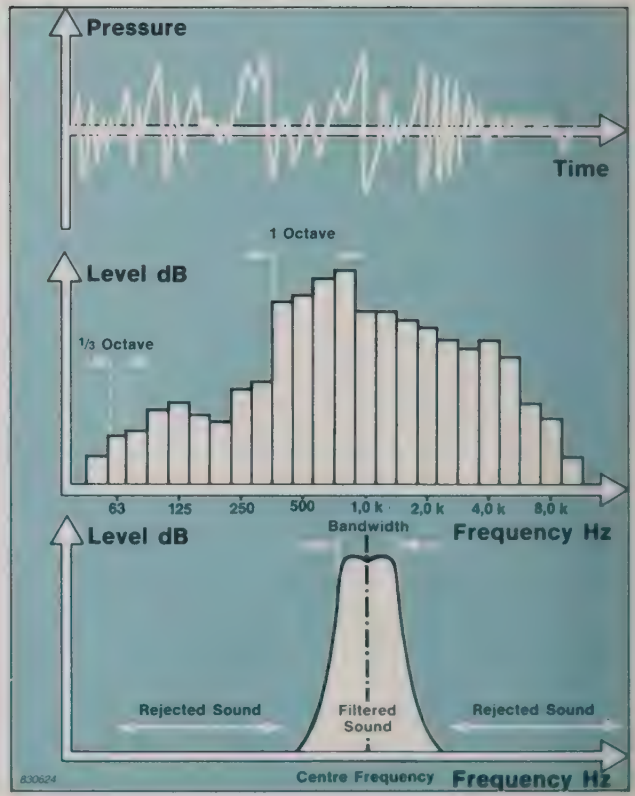


When more detailed information about a complex sound is required, the frequency range from 20 Hz to 20 kHz can be divided up into sections or **bands**. This is done with electronic filters which reject all sound with frequencies outside the selected band. These bands usually have a **bandwidth** of either **one octave** or **one third octave**.

An octave is a frequency band where the highest frequency is twice the lowest frequency. For example, an octave filter with a **centre frequency** of 1 kHz admits frequencies between 707 and 1414 Hz, but rejects all others. (The name octave stems from the fact that an octave covers eight notes of the diatonic musical scale). A third octave covers a range where the highest frequency is 1,26 times the lowest frequency.

The process of thus dividing a complex sound is termed **frequency analysis** and the results are presented on a chart called a **spectrogram**.

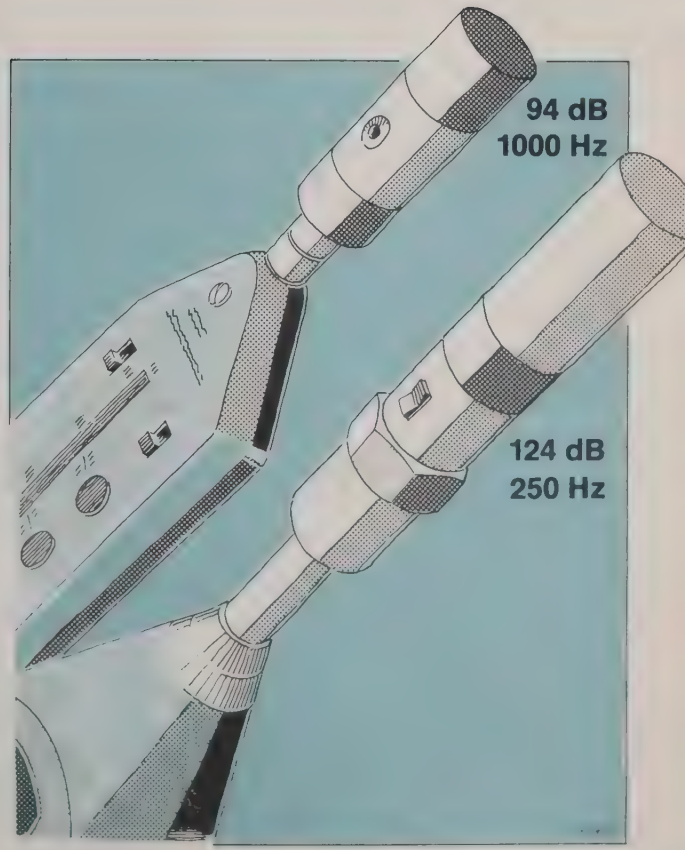
After the signal has been weighted and/or divided into frequency bands the resultant signal is amplified, and the **Root Mean Square (RMS)** value determined in an RMS detector. The RMS is a special kind of mathematical average value. It is of importance in sound measurements because the RMS value is directly related to the amount of energy in the sound being measured.



The last stage of a sound level meter is the read-out unit which displays the sound level in dB, or some other derived unit such as **dB(A)** (which means that the measured sound level has been A-weighted). The signal may also be available at output sockets, in either AC or DC form, for connection to external instruments such as level or tape recorders to provide a record and/or for further processing.

Calibration

Sound level meters should be calibrated in order to provide precise and accurate results. This is best done by placing a portable acoustic calibrator, such as a sound level calibrator or a pistonphone, directly over the microphone. These calibrators provide a precisely defined sound pressure level to which the sound level meter can be adjusted. It is good measurement practice to calibrate sound level meters immediately before and after each measurement session. If recordings are to be made of noise measurements, then the calibration signal should also be recorded to provide a reference level on playback.

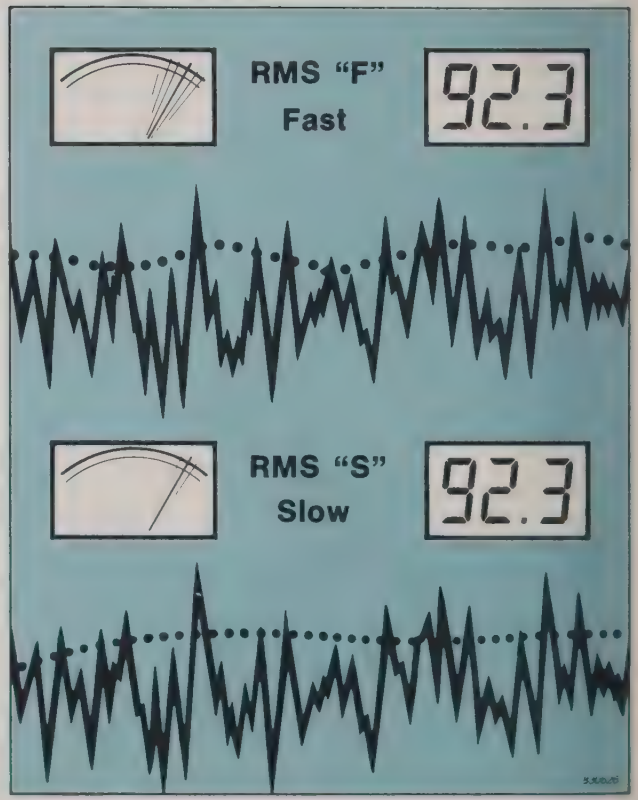


Detector Response

Most sounds that need to be measured fluctuate in level. To measure the sound properly we want to be able to measure these variations as accurately as possible. However, if the sound level fluctuates too rapidly, analogue displays (such as a moving coil meter) change so erratically that it is impossible to get a meaningful reading. For this reason, two detector response characteristics were standardized. These are known as **"F"** (for Fast) and **"S"** (for Slow).

"F" has a time constant of 125 milliseconds and provides a fast reacting display response enabling us to follow and measure not too rapidly fluctuating sound levels. "S" with a time constant of 1 second gives a slower response which helps average-out the display fluctuations on an analogue meter, which would otherwise be impossible to read using the "F" time constant.

Many modern sound level meters have digital displays which largely overcome the problem of fluctuating displays, by indicating the maximum RMS value measured within the preceding second. Selection of the appropriate detector characteristic is then often dictated by the standard upon which the measurements are to be based.



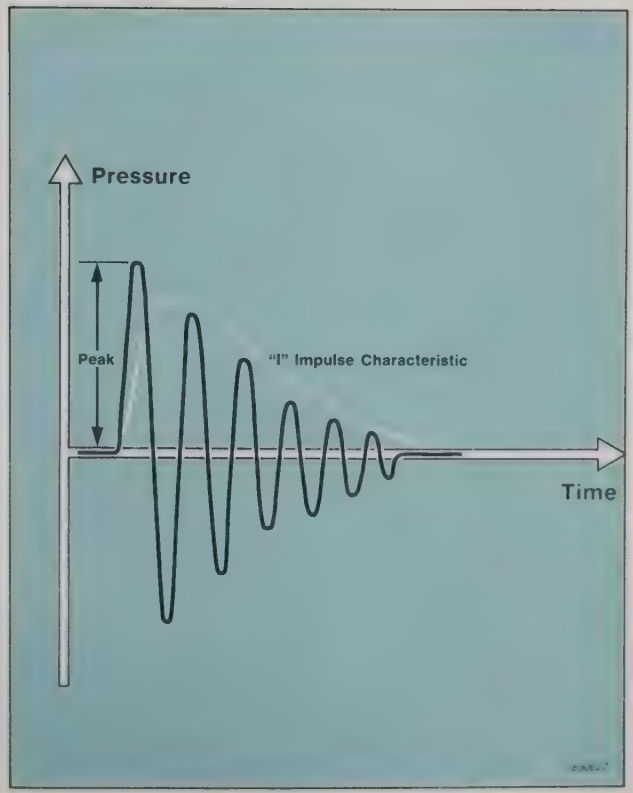
The Impulse Sound Level Meter

If the sound to be measured consists of isolated impulses or contains a high proportion of impact noise, then the normal "F" and "S" time responses of the simple sound level meter are not sufficiently short to give a measurement which is representative of the subjective human response.

For such measurements, sound level meters having a standardized **"I"** (Impulse) characteristic are needed. The "I" characteristic has a time constant of 35 milliseconds, which is short enough to enable detection and display of **transient noise**, in a way which takes into account the human perception of impulsive sounds.

Although the perceived loudness of short duration sound is lower than that of steady continuous sound, the risk of damage to hearing is not necessarily reduced. For this reason, some sound level meters include a circuit for measuring the **peak** value of the sound, independent of it's duration.

A Hold Circuit is also incorporated to store either the peak value or the maximum RMS value. Some standards require the peak value to be measured while others ask for a measurement using the "I" time constant. In either case the Hold circuit makes reading the measurement easy.

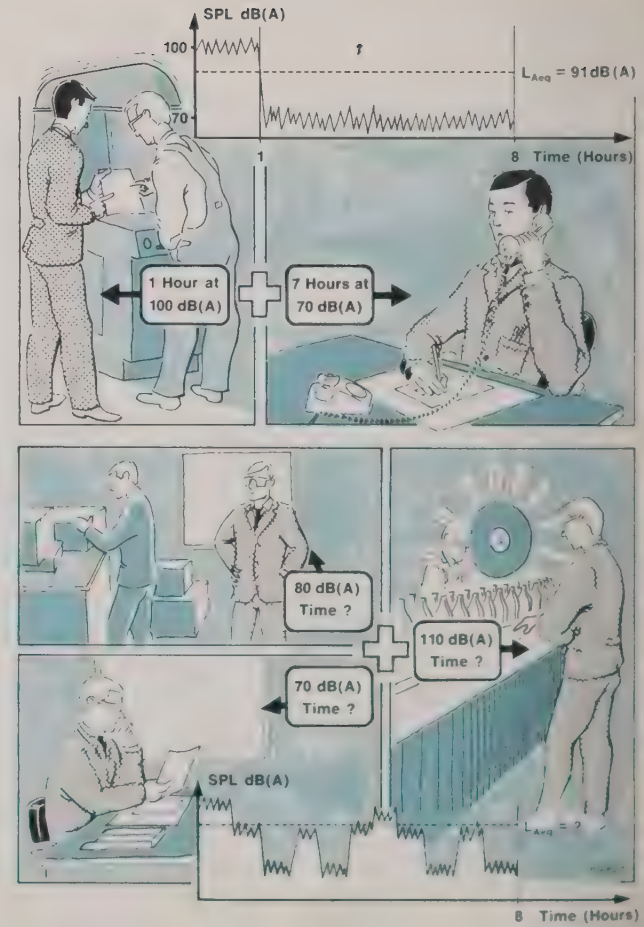


Energy Parameters

As sound is a form of **energy** the hearing damage potential of a given sound environment depends not only on it's level, but also it's **duration**. For example, exposure to a loud sound for 4 hours is much more harmful than a one hour exposure to the same sound. So to assess the hearing damage potential of a sound environment, both the sound level and the duration of exposure must be measured and combined to provide a determination of the energy received.

For constant sound levels, this is easy, but if the sound level varies, the level must be sampled repeatedly over a well defined **sampling period**. Based on these samples, it is then possible to calculate a single value known as the **Equivalent Continuous Sound Level** or L_{eq} which has **the same energy content and consequently the same hearing damage potential as the varying sound level**. For an A-weighted L_{eq} the symbol L_{Aeq} is used. In addition to determining the hearing damage potential of a sound, L_{eq} measurements are also used for many other types of noise measurements, for example community noise-annoyance assessments.

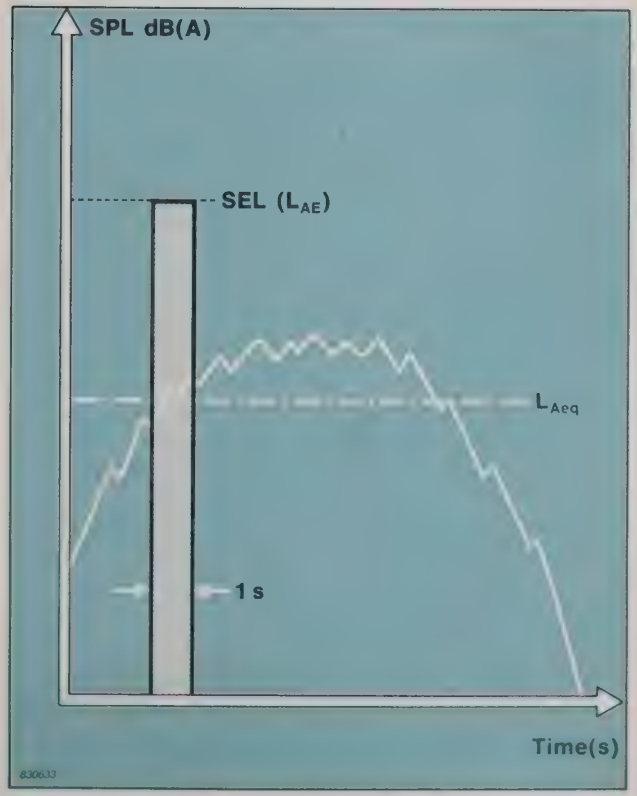
If the sound level varies in a stepwise manner, an L_{eq} can be calculated using measurements from a sound level meter and a stopwatch.



However, if the sound level varies randomly, it is not easy to use manual calculations to evaluate an L_{eq}. In such cases **Integrating Sound Level Meters** are used. Such meters automatically calculate the L_{eq} of the sound being measured. L_{eq} measurements can be performed over any suitable time period. Some sound level meters also have a 60 second L_{eq} which can be used for short term L_{eq} sampling.

An alternative energy parameter to the L_{eq} is the **Sound Exposure Level** or **SEL** which is defined as the **constant level acting for one second which has the same amount of acoustic energy as the original sound**. If based on A-weighted sound the SEL is denoted by the symbol $\mathbf{L_{AE}}$. An SEL measurement is often used for describing the noise energy of a single event, such as a vehicle passing by, or an aircraft flying over. As all SEL measurements are normalised to a one second time interval, the energy content of different types of noise events can easily be compared by using SEL measurements.

Where more information on the level distribution of the sound levels is desired, **Noise Level Analyzers** can be employed. These analyzers can provide statistical analyses of variations in the sound level over the period of interest.



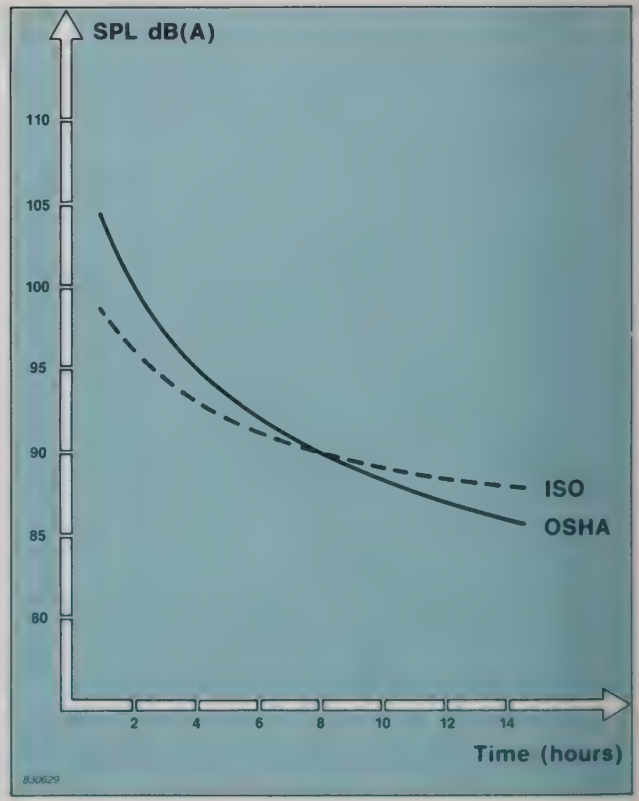
Noise Dose

Noise exposure measurements on individuals who move between many different noise environments during the working day can be obtained using **Noise Dose Meters**. These instruments are portable and can be carried in a person's pocket. The microphone can be separated from the dose meter body and should preferably be mounted close to the individuals more noise exposed ear.

Noise dose meters display the percentage of the daily allowable noise dose. Two different ways of calculating the noise dose are used. The difference between the two methods is due to the allowance incorporated for the recovery of hearing during quiet periods. Currently, both methods essentially use a basis of 90 dB(A) for an 8 hour day.

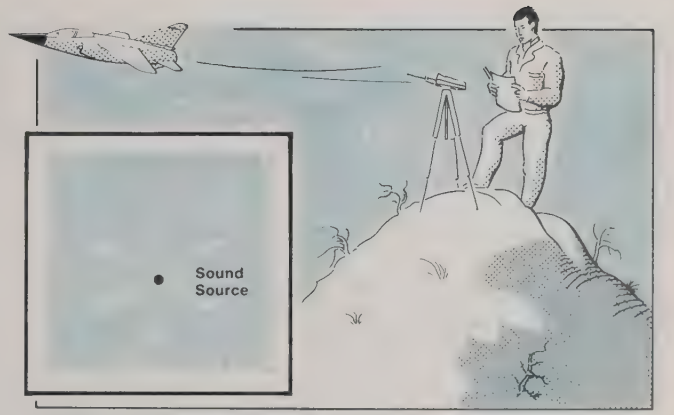
The International Standards Organisation **(ISO) 1999** defines one method which uses only the energy criteria and makes no allowance for the recovery of hearing. Thus, an increase of 3 dB in the sound pressure level halves the permissible exposure period. For example an increase in sound level from 90 dB(A) to 93 dB(A) must be accompanied by a halving of the permissible exposure duration from 8 hours to 4 hours.

In the United States the Occupational Safety and Health Administration **(OSHA)** defines another relationship which permits a 5 dB increase in sound level for each halving of the allowable exposure period. Thus, an increase in sound level from 90 dB(A) to 95 dB(A) is accompanied by a halving of the allowable exposure duration from 8 to 4 hours.



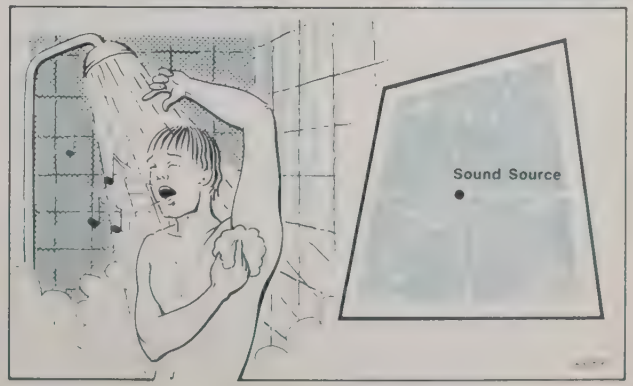
Anechoic Chambers (Sound Absorbing Rooms)

In order to make measurements in a free-field, totally without reflecting objects, the measurements must be made outdoors at the top of a flagpole (or equivalent) or in an **anechoic** chamber. In an anechoic chamber the ceiling, floor and all the walls are covered by a highly absorptive material which eliminates reflections. Thus, the sound pressure level in any given direction from the noise source may be measured without the presence of interfering reflections.



Reverberation Chambers (Sound Reflecting Rooms)

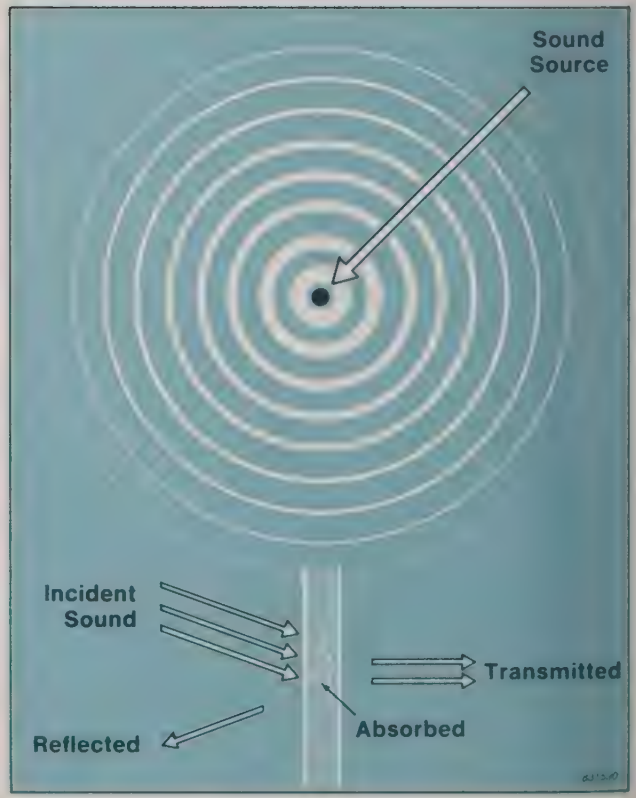
The opposite of an anechoic chamber is the **reverberation** chamber where all surfaces are made as hard and reflective as possible and where no parallel surfaces exist. This creates a so-called **diffuse** field because the sound energy is uniformly distributed throughout the room. In this type of room, it is possible to measure the total acoustic power output from the noise source, but the sound pressure level at any point will be an average value due to the reflections. As such rooms are cheaper to construct than anechoic chambers, they find widespread use for machinery noise investigations.



Sound Propagation

Sound propagation in air can be compared to ripples on a pond. The ripples spread out uniformly in all directions, decreasing in amplitude as they move further from the source. For sound in air, when the distance doubles, the amplitude drops by half — which is a drop of 6 dB. Thus, if you are at a position one meter from the source and move one meter further away from the source, the sound pressure level will drop by 6 dB. If you move to 4 meters, it will drop by 12 dB, 8 meters by 18 dB, and so on. However, this is only true when there are no reflecting or blocking objects in the sound path. Such ideal conditions are termed **free-field** conditions.

With an obstacle in the sound path, part of the sound will be **reflected,** part **absorbed** and the remainder will be **transmitted** through the object. How much sound is reflected, absorbed or transmitted depends on the properties of the object, its size and the wavelength of the sound. In general, the object must be larger than one wavelength in order to significantly disturb the sound. For example, at 10 kHz the wavelength is 3,4 cm — so even a small object such as a measurement microphone will disturb the sound field — hence sound absorption and insulation are readily achieved. But, at 100 Hz, the wavelength is 3,4 meters and sound insulation becomes much more difficult. You've probably noticed this with music playing in the room next door. It is the bass which is very difficult to block out.

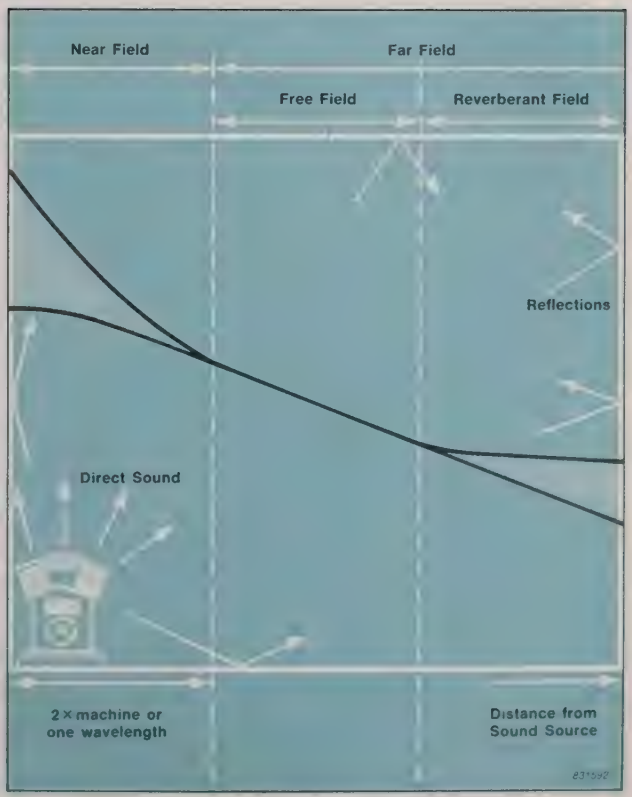


The Practical Room

In practice the majority of sound measurements are made in rooms that are neither anechoic nor reverberant — but somewhere in-between. This makes it difficult to find the correct measuring positions when the noise emission from a given source must be measured.

When determining emission from a single source, several errors are possible. If measurements are made too close to the machine, the SPL may vary significantly with a small change in sound level meter position. This will occur at a distance less than the wavelength of the lowest frequency emitted from the machine, or at less than twice the greatest dimension of the machine, whichever distance is the greater. This area is termed the **near-field** of the machine, and measurements in this region should be avoided if possible.

Other errors may arise if you measure too far away from the machine. Here, reflection from walls and other objects may be just as strong as the direct sound from the machine and correct measurements will not be possible. This region is termed the **reverberant-field.** Between the reverberant and near-field is the free-field which can be found by noting that the level drops 6 dB for each doubling in distance from the source. SPL measurements should be made in this region. However, it is quite possible, that the conditions are so reverberant or the room is so small that no free-field exists. In such cases some standards (such as ISO 3746) suggest an environmental correction to account for the effect of reflected sound.



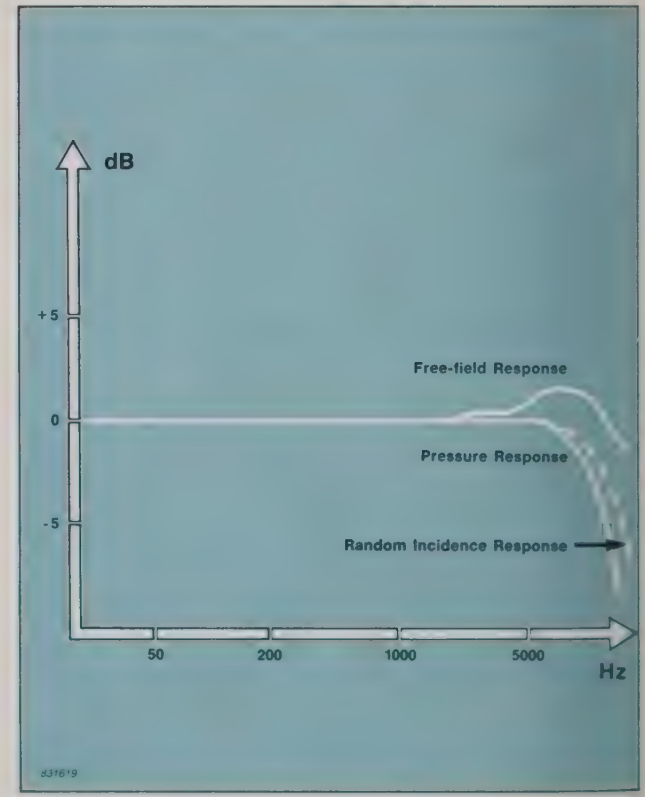
The Microphone in the Sound Field

The type of microphone and its orientation in the sound field also influence the accuracy of measurements. A measurement microphone should have a uniform frequency response, that is the microphone must be equally sensitive throughout the frequency range.

A microphone is normally characterized by one of three types of frequency response characteristics — **free-field** (usually at 0° incidence), **pressure,** and **random-incidence**, and is named after the response that is the most linear. Thus, the response curves shown in the diagram are for a random incidence microphone.

It is important to note that any microphone will disturb a sound field, but the free-field microphone compensates for the disturbance it causes in the sound field. The pressure microphone however, responds uniformly to the actual SPL, including the pressure disturbance caused by the microphone itself. The random incidence microphone is designed to respond uniformly to sounds arriving simultaneously from all angles, as is the case in highly reverberent or diffuse sound fields. (For most microphones the pressure and random incidence responses are very similar so a pressure microphone may also be used for random incidence measurements).

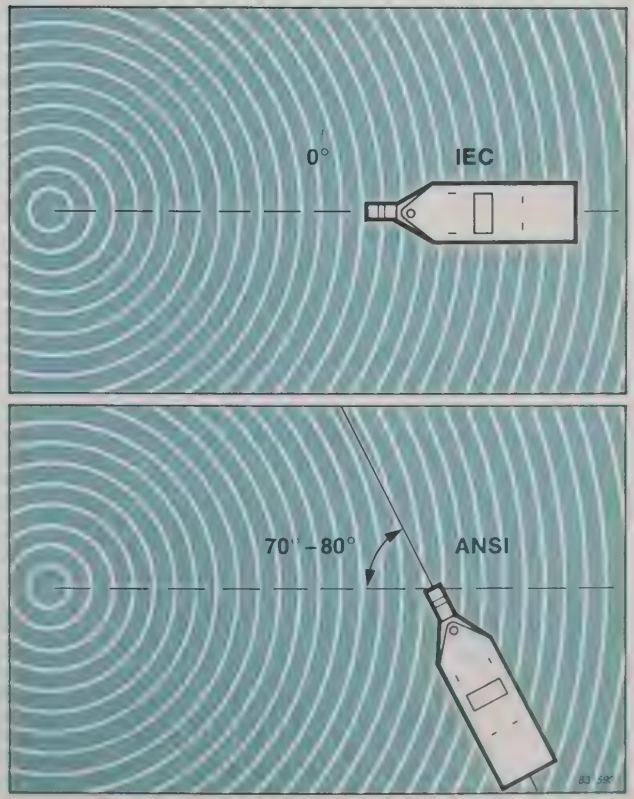
In general, when making free-field measurements (most outdoor measurements are essentially free-field), use a free-field microphone. In a diffuse-field, the microphone should be as omnidirectional as possible.



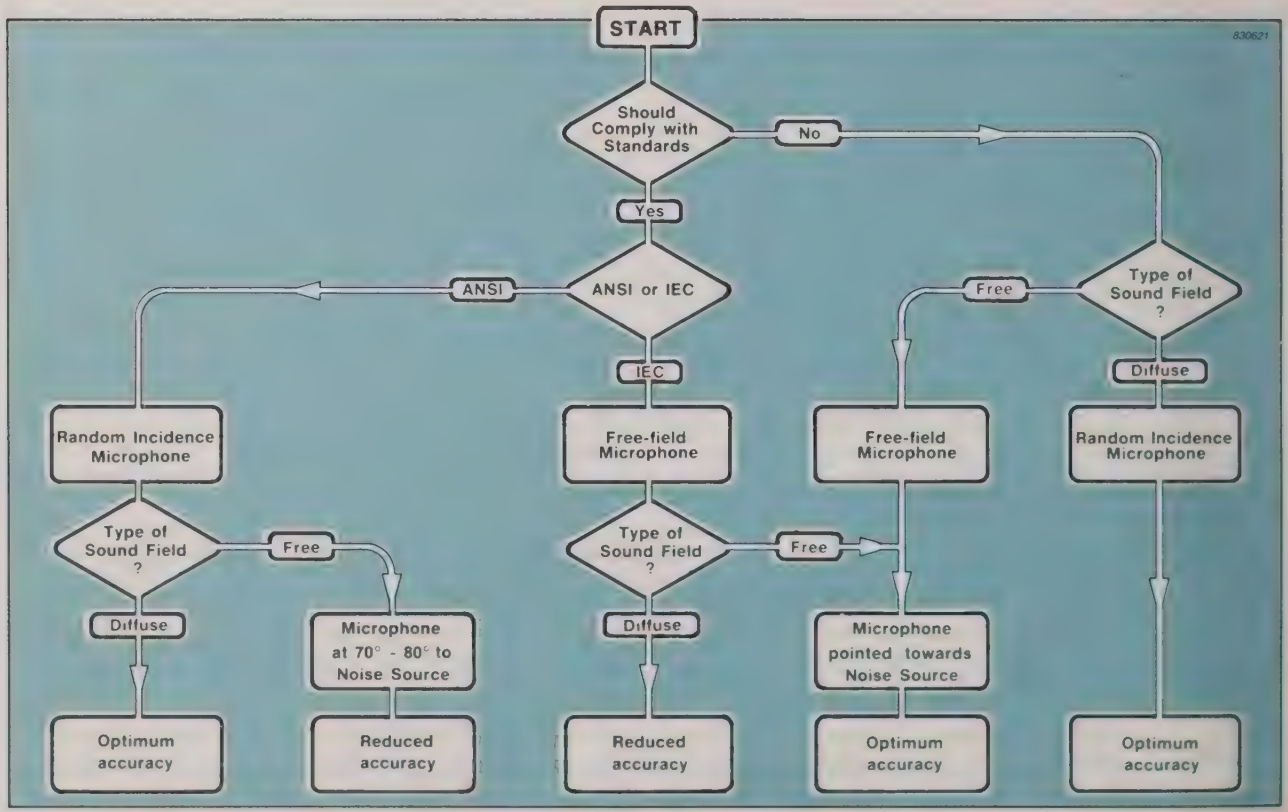
Selection of the most appropriate microphone may also be influenced by applicable national or international standards. For example the International Electrotechnical Commission **(IEC)** specifies sound level meters with a free-field response, whilst the American National Standard **(ANSI)** calls for a random incidence response to be used.

If a random-incidence microphone is used in a free-field environment, the most accurate measurement is obtained when it is oriented at an angle of between 70° and 80° to the sound source. If pointed directly at the source, the resulting measurement will be too high. Conversely, a free-field microphone used in a diffuse field under-estimates the true SPL.

The response of a free-field microphone can be changed by fitting an acoustic resonator. The resonator increases the pressure in front of the microphone at high frequencies, modifying the response to one that is closer to a flat random response. Such a resonator cap should be used when measuring indoors (i.e. in a predominantly diffuse field). In other cases and when measuring outdoors, remove the resonator cap and point the microphone towards the source. Some sound level meters can change from a correct free-field response to a very accurate random-incidence response by simply switching-in specially incorporated circuits.



The Microphone in the Sound Field



Influence of Instrument and Operator

When measuring noise, precautions should be taken to keep the sound level meter and operator from interfering with the measurement. Not only can the instrument body and operator's presence block sound coming from a given direction, but they can cause reflections that may cause measurement errors. You may never have thought of your body as a sound reflector, but experiments have shown that at frequencies of around 400 Hz, reflections from a person may cause errors of up to 6 dB when measuring less than one meter from the person.

To minimize the reflections from the instrument body, all Brüel & Kjær sound level meters are specially designed with a conically shaped front end. For even more precise measurements, some instruments are provided with an extension rod for mounting the microphone away from the instrument body to avoid disturbing the sound field around the microphone. Extension cables are also available to provide a truely remote microphone. To minimize the reflection of sound from your body, it is usually sufficient to hold the sound level meter at arms length. It may also be mounted on a tripod, possibly fitted with it's extension rod. You can check whether your presence is influencing the meter reading by letting the sound level meter remain fixed while you step from side to side.

The Influence of the Environment

Wind

Wind blowing across the microphone produces a lot of extraneous noise, similar to the noise you can hear with the wind blowing in your ear. To reduce this noise, a special windscreen consisting of a ball of porous sponge should always be used over the microphone. It also shields the microphone from dust, dirt and precipitation, and helps to protect it from mechanical damage.

Humidity

In most cases relative humidity levels up to 90 % will have a negligible effect on the sound level meter and microphone. However, care should be taken to shield the instrument from rain, snow, etc. A windscreen should always be fitted over the microphone during precipitation. Even if the windscreen becomes very wet, measurements will still be accurate. However, for continuous use in extremely humid environments, special outdoor microphones, rain covers and dehumidifiers are recommended.

Temperature

All Brüel & Kjær sound level meters are designed to operate accurately over the –10 to +50 °C (14 to 122 °F) range. However, care should be taken to avoid sudden temperature changes which may lead to condensation in the microphone.

Ambient Pressure

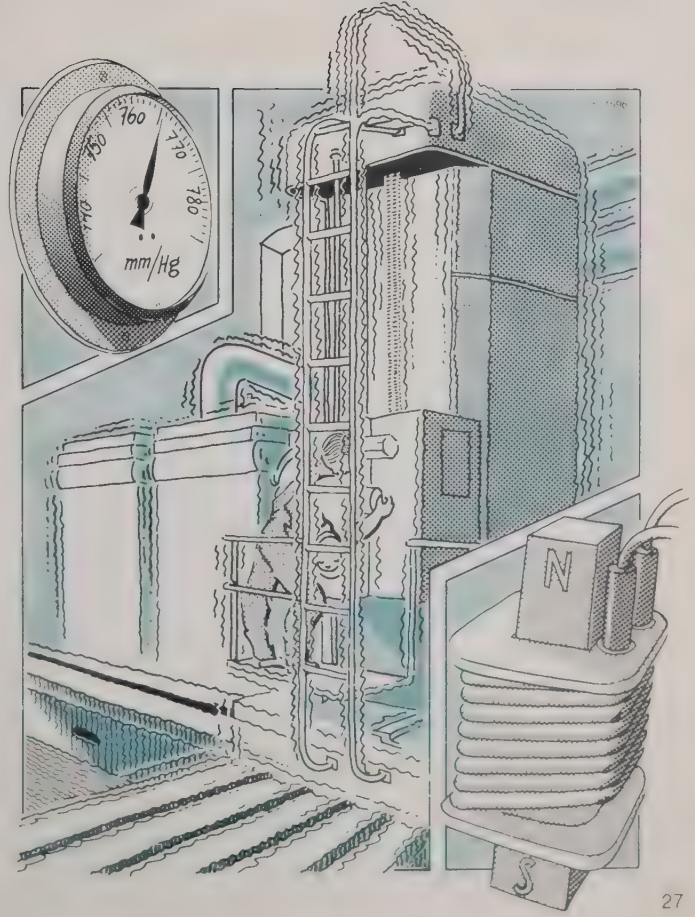
Variations in atmospheric pressure of ±10 % will have a negligible influence (less than ±0,2 dB) on microphone sensitivity. However, at high altitudes the sensitivity may be affected by more than this, especially at high frequencies, and the microphone instruction manual should be consulted. Also, when calibrating the instrument with a Pistonphone, a correction must be made for atmospheric pressure.

Vibration

Although the microphone and sound level meter are relatively insensitive to vibration, it is always a good practice to isolate them from strong vibrations and shock. Foam rubber pads or similar isolating material may be used if the sound level meter must be used in a high vibration environment.

Magnetic Fields

The influence of electrostatic and magnetic fields on sound level meters is negligible.



Environmental Noise

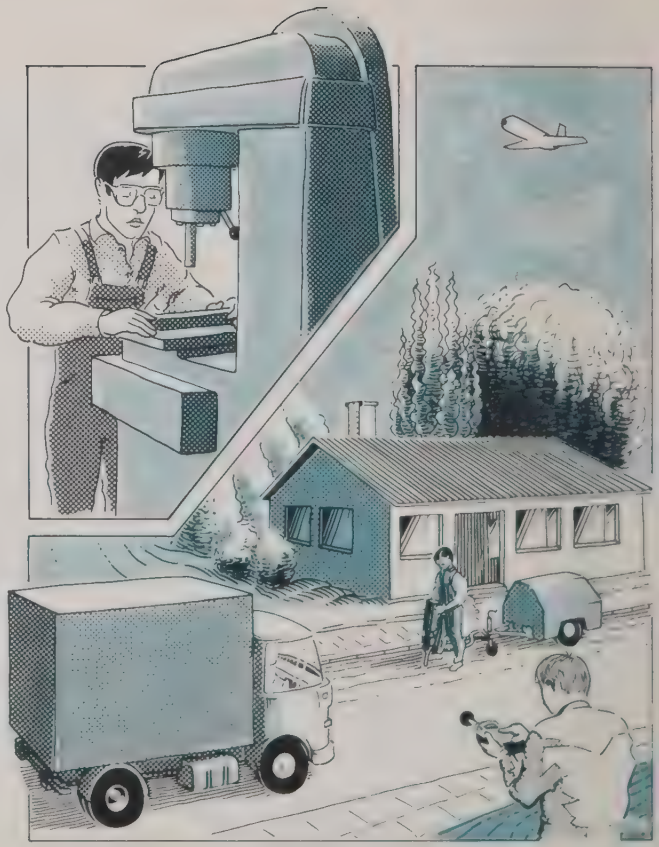
So far we have only been interested in the measurement of noise emitted by a single source, for example to rate a machine in order to predict its noise at greater distances.

However, **environmental noise** involves measurement of the total noise (irrespective of its source) at a particular location. The noise may thus be due to one or more sources and may also include reflections from walls, ceilings and other machines.

Noise at an employees work station is an example of environmental noise. The measurement is made where the person works, without regard to whether they are in the near or far field of their machine or whether other machines are operating nearby. These conditions may be considered in efforts to reduce the noise level, but not when measuring the employee's actual exposure. They may also mean that one ear of the operator receives more noise energy than the other.

Because environmental sounds come from various directions, the sound level meter should be omnidirectional. It must have a uniform response irrespective of where the various sound sources are located.

Other instances where environmental noise measurements are used are in the community (e.g. road traffic noise measurements), at factory boundaries, in offices and in theatres.



Background Noise (Subtracting Sound Levels)

One factor that may influence the accuracy of measurements is the level of the **background noise** compared to the level of the sound being measured. Obviously, the background noise must not "drown out" the sound of interest. In practice, this means that the level of the sound must be at least 3 dB higher than the background noise. However, a correction may still be necessary to get the correct result. The procedure for measuring the sound level from a machine under conditions of background noise is as follows:

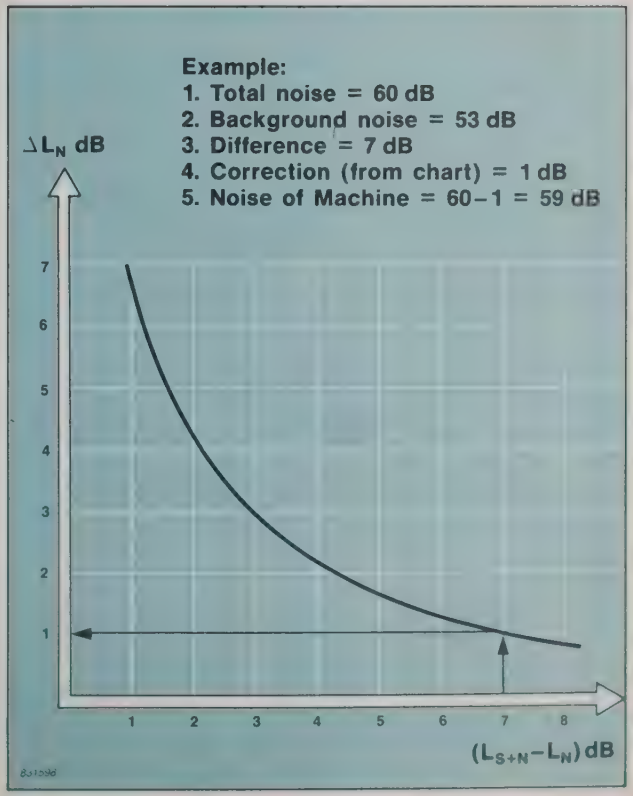
1. Measure the total noise level (L_{S+N}) with the machine running.

2. Measure the background noise level (L_N) with the machine turned off.

3. Find the difference between the two readings ($L_{S+N} - L_N$). If this is less than 3 dB, the background noise level is too high for an accurate measurement. If it is between 3 and 10 dB, a correction will be necessary. No correction is necessary if the difference is greater than 10 dB.

4. To make corrections, the chart shown may be used. Enter the bottom of the chart with the difference value ($L_{S+N} - L_N$) from step 3, go up until you intersect the curve and then go to the vertical axis on the left.

5. Subtract the value on the vertical axis (ΔL_N) from the total noise level in step 1. This gives the sound level L_S of the machine.



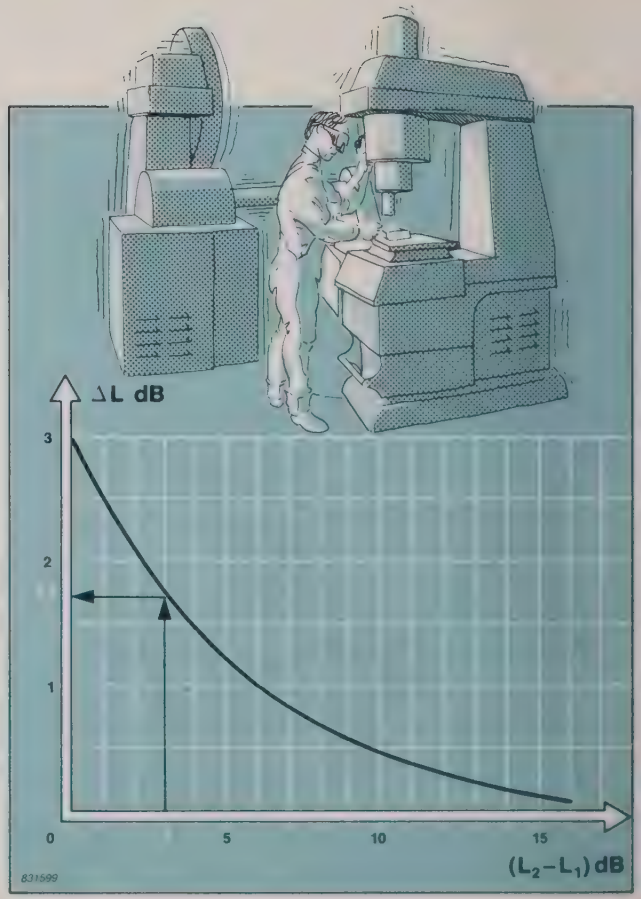
Addition of Sound Levels

If the sound levels from two or more machines have been measured separately and you want to know the total SPL made by the machines when operating together, the sound levels must be added. However, dBs cannot just be added together directly (because of the logarithmic scale). Addition of dBs can be done simply using the chart opposite and the following procedure:

1. Measure the SPL of each machine separately (L_1, L_2).
2. Find the difference between these levels ($L_2 - L_1$).
3. Enter the bottom of the chart with this difference. Go up until you intersect the curve, then go to the vertical axis on the left.
4. Add the value indicated (ΔL) on the vertical axis to the level of the noisier machine (L_2), This gives the sum of the SPLs of the two machines.
5. If 3 machines are present, repeat steps 1 to 4 using the sum obtained for the first two machines and the SPL for machine three.

Example:

1. Machine 1		L_1	= 82 dB
Machine 2		L_2	= 85 dB
2. Difference	$L_2 - L_1$		= 3 dB
3. Correction (from chart)		ΔL	= 1,7 dB
4. Total Noise	= 85 + 1,7		= 86,7 dB



Standardization of Measurements

Whenever sound measurements are made, the recommendations of the applicable national and international standards should be studied. These standards discuss both measuring techniques and specifications for the equipment used. The standards provide well defined procedures for making accurate and reproducible measurements.

"Acoustics — Guide to International Standards on the measurement of airborne acoustical noise and evaluation of its effects on human beings" is the title of **ISO 2204** which should be of particular interest to the newcomer. It defines basic terms and measuring methods and also gives a reference list of other applicable standards.

IEC 651 is a standard established by the International Electrotechnical Commission which defines the specifications for various grades of sound level meters. All Brüel & Kjær sound level meters are built to conform to this standard. In the United States and some other countries, the American National Standard **ANSI S1.4-1983** is used. Brüel & Kjær sound level meters also conform to the ANSI standard when switched to random incidence mode or used with the random incidence adaptor provided.

Brüel & Kjær also produce two booklets which may be of interest. These are called "Acoustic Measurements According to ISO Standards and Recommendations" and "National and International Standards and Recommendations — Acoustics, Vibration & Shock, Luminance & Contrast". To obtain these, please contact our local representative.

The Measurement Report

A very important part of sound measurements is careful documentation of the measurements and results. A good measurement report should contain at least the following information:

1. A sketch of the measurement site showing applicable dimensions (e.g. size of room, machine dimensions), the location of the microphone and object being measured.

2. Standard(s) to which measurements are made.

3. Type and serial number of instrument(s) used.

4. Method of calibration.

5. Weighting networks and detector responses used.

6. Description of type of sound (e.g. impulsive, continuous, tones etc.)

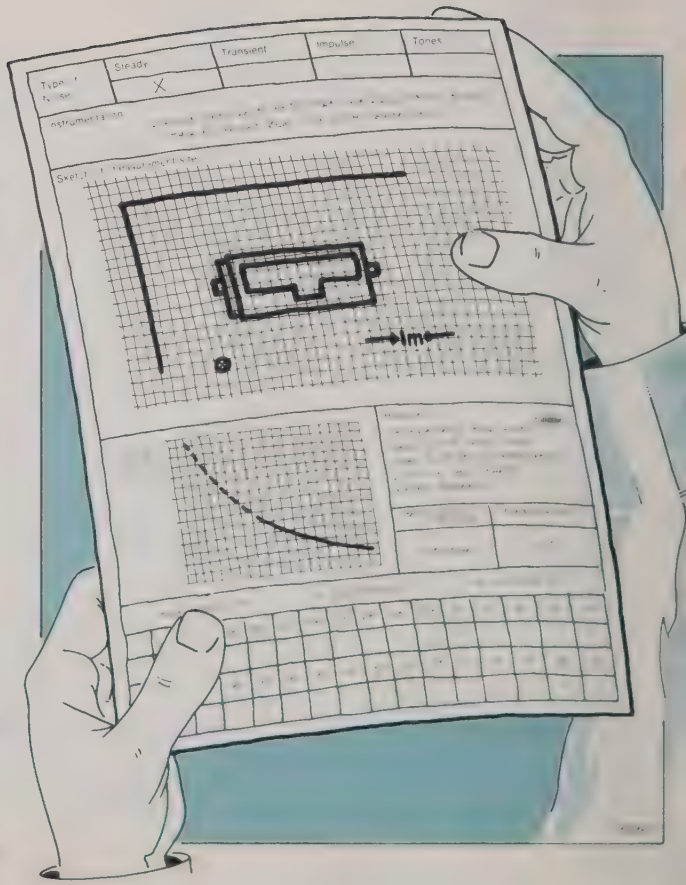
7. Background noise level.

8. Environmental conditions (e.g. type of sound field, atmospheric conditions).

9. Data on object being measured (e.g. machine type, load, speed etc.)

10. Date when measurements were performed.

With a carefully written report, future comparisons will be more accurate and reliable.



Noise Mapping

One of the first steps in a noise abatement programme will usually be to make a **noise map.** A reasonably accurate sketch showing the relative positions of all machines and other items of interest is drawn. Sound level measurements are obtained at several positions around the area being investigated and the positions are plotted on the sketch. Connecting lines are drawn between points with equal sound levels, thus producing **noise contours** which indicate the sound distribution pattern; the more measurements, the more accurate the noise map.

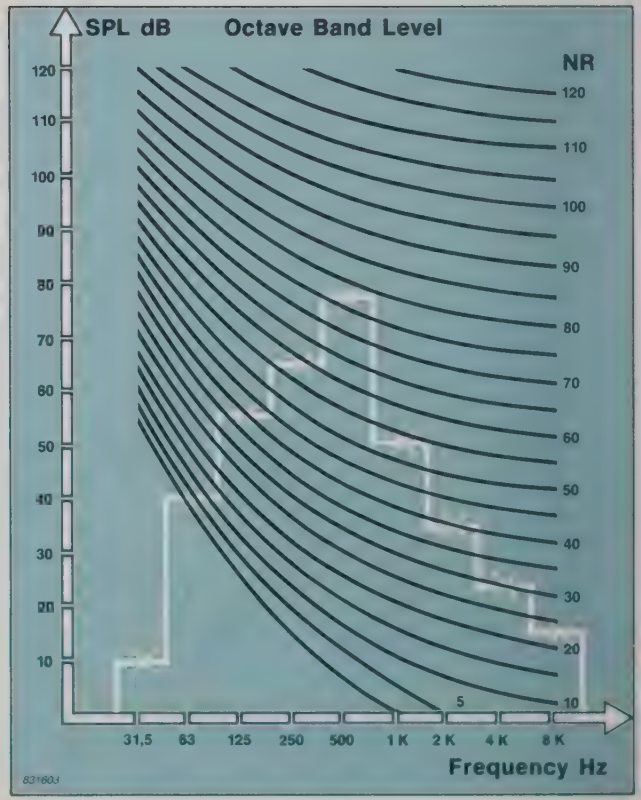
A map of this kind will immediately show in which areas the noise levels are too high. This provides a starting point for planning the steps to be taken to protect the workers. When the necessary sound reduction measures have been made, new measurements will give a clear picture as to what extent the noise patterns have been changed. A noise map could also be used to indicate areas where the wearing of ear protection is obligatory.

Using Noise Rating Curves

In almost any noise abatement programme, in particular when the measured dB(A) values are above an allowable limit, it will be necessary to try to evaluate the annoyance value of the noise. For this purpose, octave or third octave band frequency analyses must first be obtained.

Several methods (some quite complicated) for determining the level of noise annoyance are recommended in various standards. The simplest method is to use rating curves such as the **Noise Rating (NR) Curves** given in **ISO R 1996** which are shown on the right. The NR curves are used by plotting the octave band noise spectrogram over them. It can then be seen which NR curve lies just above the spectrogram and the noise is then assigned that particular NR number (in the example, NR 78).

From the shape of the curves it can be seen that much more importance is given to the higher frequencies than the lower ones. This is because high frequency sounds are more annoying than lower frequency sounds. In some countries, curves similar to NR curves are used to give the maximum allowable time that the worker may be exposed to the noise, as well as being used to set allowable limits for machinery etc.



When Levels are too High

When measurements have shown that the sound levels are too high, steps must be taken to reduce them. Although the details of a noise abatement program can be quite complex, there are four general guidelines to possible solutions:

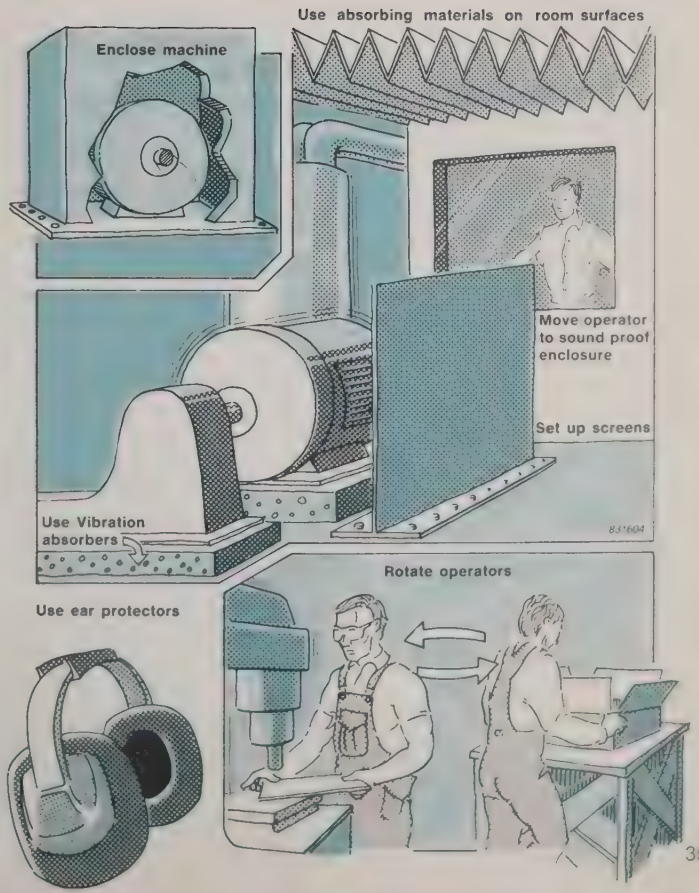
1. Eliminate or reduce the noise at its source. This may be done by acoustic treatment of machine surfaces, redesign of the machine, or purchasing a new quieter machine.

2. Block the sound transmission path, for example by placing an enclosure or acoustic screens around the machine and mounting it on vibration isolators to prevent transmission through the floor. Noise is further reduced by coating walls, ceiling and floor with absorbent materials to reduce reflections from their surfaces. Alternatively, the operator(s) may be provided with **noise refuges** or enclosures.

3. Provide the exposed person with hearing protection. However, this should not generally be regarded as a permanent solution.

4. Shut down the offending machinery. In severe cases, this step must be considered. It is also possible to limit the hours of operation, move the offending noise source to another location, or rotate machine operators so their exposure time is reduced.

Further information about methods of reducing high noise levels can be found in the Brüel & Kjær booklet **"Noise Control. Principles and Practice."**

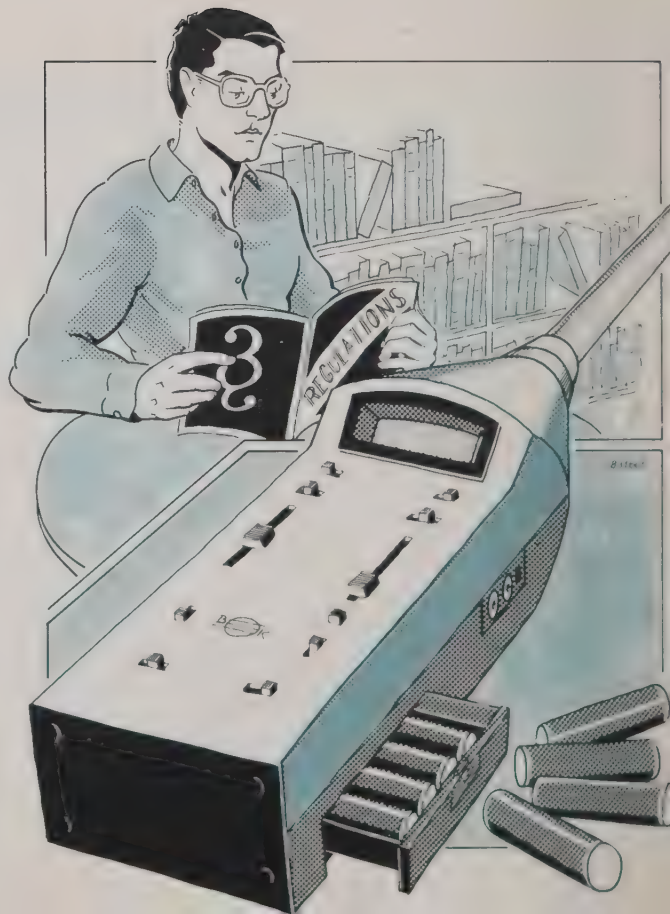


Some Basic Rules

To conclude this booklet, let us review some basic rules to follow when making sound measurements using a portable sound level meter.

1. Check applicable standards and rules for the proper measuring equipment and techniques (see p. 31 and p. 42).

2. Check that the batteries for the measuring instruments and the calibrator are sufficiently charged and take along extra sets of good quality batteries. If instruments are to be stored for a long time, the batteries should be removed.

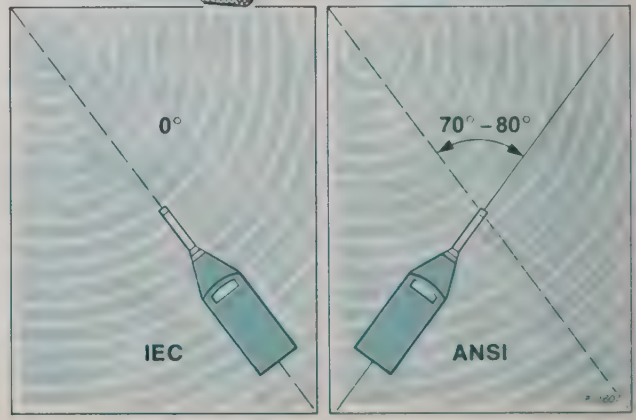


3. Make sure that the instrument is properly calibrated. Use of an acoustic calibrator at the start and end of each measurement session is recommended.

4. Make a sketch of the area and perform some orientation measurements before noting actual values. Determine the kind of sound field you are working with (see p. 22, 23 and 24) and find the correct measuring positions (see p. 21).

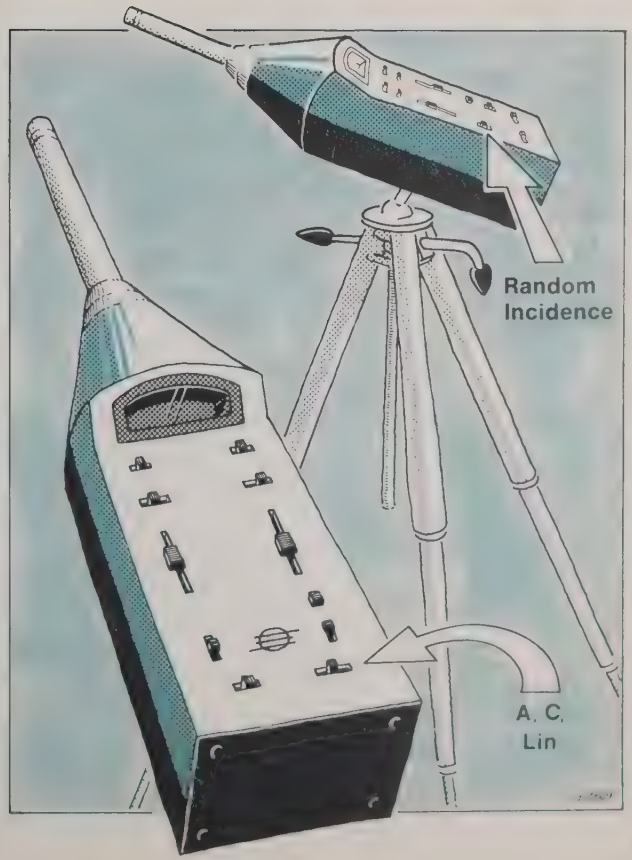
5. If you are not sure about the exact source of sound being measured, the use of a set of headphones connected to the output of the sound level meter will help you identify the sound. This is of course only possible if the SLM has an AC output.

6. If measuring in a free-field environment, point a free-field microphone (IEC) directly towards the source. If using a random incidence microphone (ANSI) orient the microphone 70 ° - 80 ° from the source (see p. 22-24).



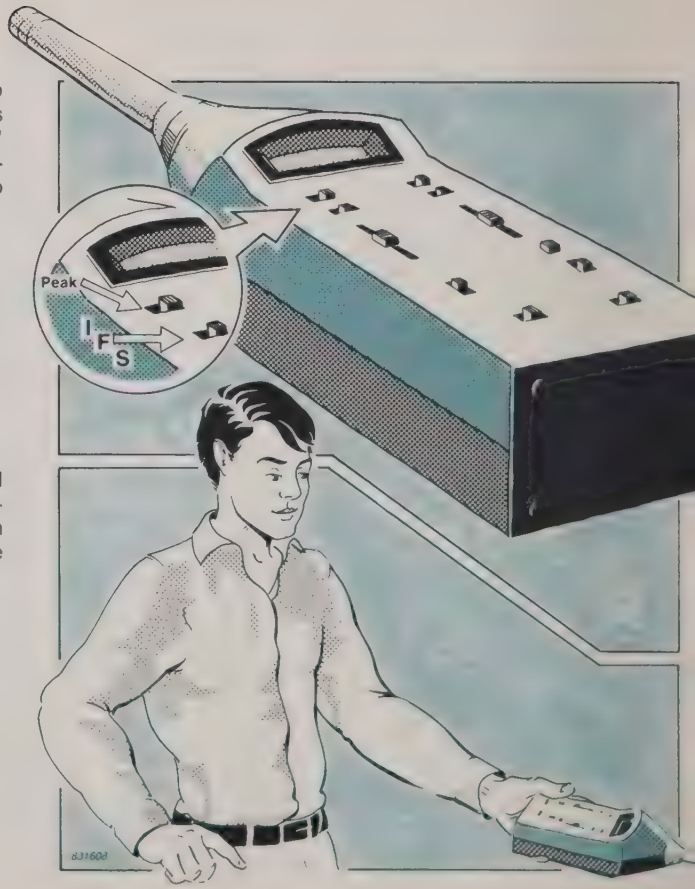
7. If the sound comes from more than one direction, it is important to choose a microphone and mounting which gives the best possible omnidirectional characteristics (see p. 22-25).

8. Decide which weighting network should be used. Normally this would be the "A" network (see p. 11), but may depend on the standards you are using.



9. Select the correct detector response, "F" or "S" to get an accurate reading (see p. 14). If the sound is impulsive, an "Impulse Sound Level Meter" with "I" response should be used (see p. 15). If required, energy measurements such as L_{eq} and SEL should also be made (see p. 16 and 17).

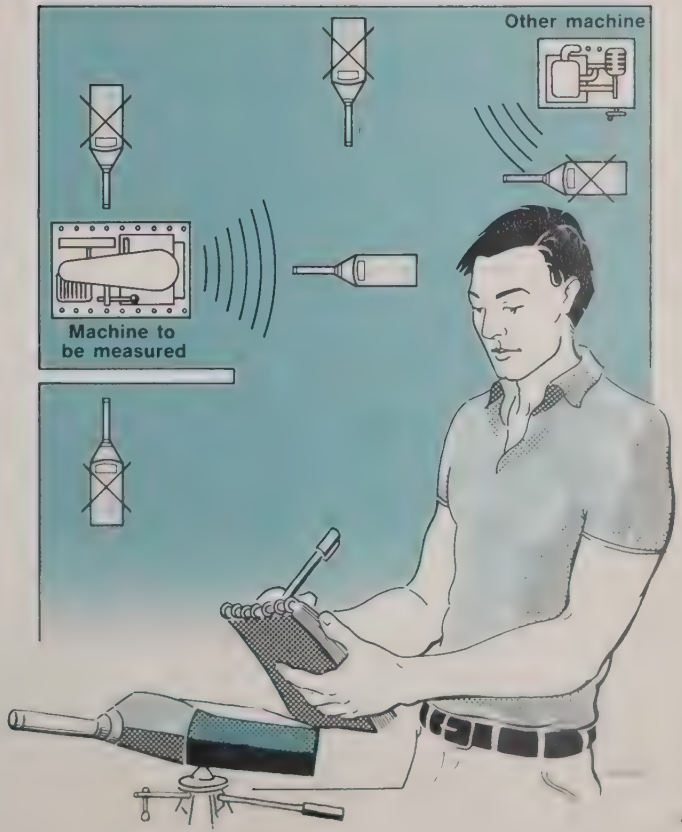
10. When making sound measurements, hold the sound level meter at arms length or use a remote microphone. This will help to avoid both reflections from your body and also blocking of sound from some directions (see p. 25).



11. During the measurement, remember to:

a. Keep away from reflecting surfaces.
b. Measure at a suitable distance from the noise source (see p. 21).
c. Check the background noise level (see p. 29).
d. Make sure that nothing obstructs the noise source.
e. Use a windshield.
f. Be careful not to accept readings if the meter is overloaded.

12. Keep a well documented measurement report (see p. 32).



Further Reading

ANON.	Acoustics Measurements according to ISO Standards and Recommendations, Brüel & Kjær, 1980
ANON.	National and International Standards and Recommendations — Acoustics, Vibration & Shock, Luminance & Contrast, Brüel & Kjær,1981
ANON.	Noise Control. Principles and Practice, Brüel & Kjær, 1982
A.N.S.I.	Specification for Personal Noise Dosemeters, Publication S1.25, 1978
A.N.S.I.	Specification for sound level meters, Publication S1.4, 1983
HASSALL & ZAVERI	Acoustic Noise Measurements. Brüel & Kjær, 1979
I.E.C.	Sound Level Meters, Publication 651, 1979
I.S.O.	Acoustics - Assessment of noise with respect to community response. Publication R 1996, 1971.
I.S.O.	Acoustics - Assesssment of occupational noise exposure for hearing conservation purposes. Publication 1999, 1975
I.S.O.	Acoustics - Determination of sound power levels of noise sources - Survey method. Publication 3746, 1979
I.S.O.	Acoustics - Guide to International Standards on the measurement of airborne acoustical noise and evaluation of its effects on human beings, Publication 2204, 1979
I.S.O	Acoustics - Description and measurement of environmental noise - Part 1: Basic quantities and procedures, Publication 1996/1, 1982

We hope this booklet has served as an informative introduction to the measurement of sound and will continue to serve as a handy reference guide. If you have other questions about measurement techniques or instrumentation, please contact one of our local representatives or write to :

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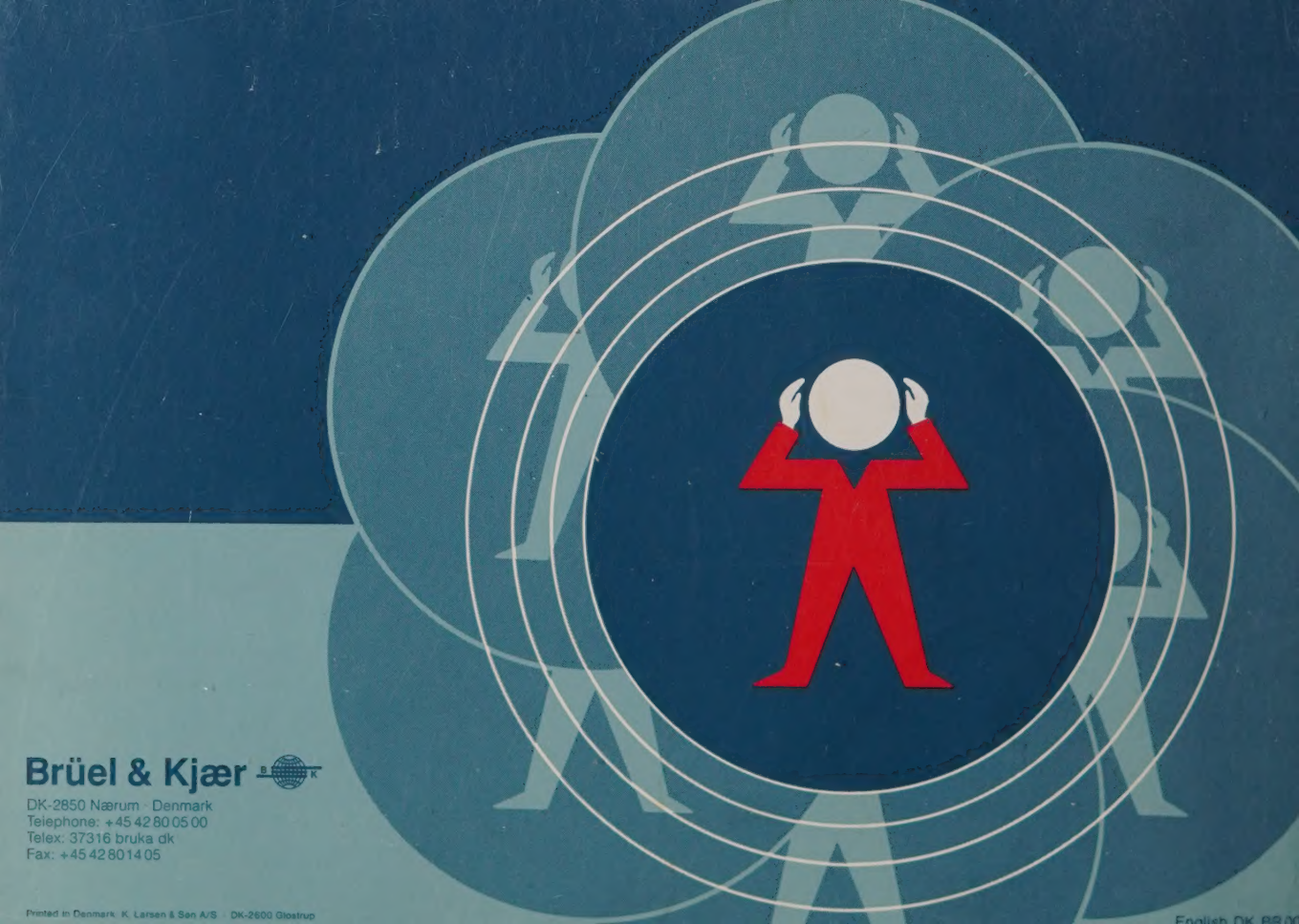
2850 Nærum
Denmark



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